Q77683.APP.CWB
October 23, 2003

METHOD OF ENCODING AND/OR DECODING DIGITAL AUDIO USING TIME-FREQUENCY CORRELATION AND APPARATUS PERFORMING THE METHOD

[01] This application claims priority from Korean Patent Application No. 02-82380, filed December 23, 2002, the contents of which are incorporated herein by reference in their entirety.

BACKGROUND OF THE INVENTION

1. Field of the Invention

[02] The present invention relates to a digital audio encoding and/or decoding method and an apparatus performing the same, and more particularly, to an audio encoding and/or decoding method for improving a prior art encoding and decoding apparatus by using the time-frequency correlation of an audio signal, and apparatus thereof.

2. Description of the Related Art

[03] Audio encoders and decoders, that is, audio codecs, are widely used because they enable users to send music files through the Internet at a lower bitrate. Among audio codecs, MP3 codecs that are used to share music files through the Internet and to play music files in portable audio players have become standard. The number of MP3 music files available on the Internet and the users sharing MP3 music files are increasing exponentially.

[04] In the audio coding field a great amount of research and development has been performed in order to implement audio codecs that can compress an audio signal at a low bitrate while maintaining the original sound quality. These audio codecs include motion picture experts group (MPEG)-1 layer 3, MPEG-2 advanced audio coding (AAC), MPEG-4, and Windows Media Audio (WMA).

[05] FIG. 1 is a block diagram of a prior art MPEG audio encoding apparatus. Here, an MPEG-1 layer 3 audio encoder, that is, an MP3 audio encoder, will now be explained as an example.

[06] MP3 audio encoders comprise a filter bank 110, a fast Fourier transform (FFT) unit 120, a psychoacoustic model unit 130, a modified discrete cosine transform (MDCT) unit, and a quantization and Huffman encoding unit 150.

[07] The filter bank 110 divides an input time-domain audio signal into 32 frequency-domain subbands in order to remove the statistical redundancy of the audio signal.

[08] The FFT unit 120 converts the input audio signal into a frequency-domain spectrum and outputs the spectrum to the psychoacoustic model unit 130.

[09] In order to remove perceptual redundancy resulting from the characteristics of human hearing, by using the frequency spectrum output from the FFT unit 120, the psychoacoustic model unit 130 determines a masking threshold which is a noise level a human-being cannot perceive, that is, a

signal to mask ratio (SMR), for each subband. The SMR value determined in the psychoacoustic model unit 130 is input to the quantization and Huffman encoding unit 150.

[10] Also, the psychoacoustic model unit 130 determines whether or not to switch a window, by calculating perceptual energy, and outputs the window switching information to the MDCT unit 140.

[11] In order to increase frequency resolution, the MDCT unit 140 divides the subbands that are divided in the filter bank 110, into finer frequency bands, by using the window switching information input from the psychoacoustic model unit 130.

[12] Based on the SMR value input from the psychoacoustic model unit 140, the quantization and Huffman encoding unit 150 processes the frequency-domain data, which is input from the MDCT unit 140 after being MDCT transformed, by performing bit allocation for removing perceptual redundancy and quantization for audio signal encoding.

[13] The audio encoding method using a psychoacoustic model shown in FIG. 1 is disclosed in U.S. Patent No. 6,092,041. Since audio codecs such as the MP3 encoder shown in FIG. 1 perform encoding and decoding at low bitrates, the output audio quality is degraded.

SUMMARY OF THE INVENTION

[14] The present invention provides an audio encoding method and apparatus by which the performance of the prior art encoding apparatus is improved such that better sound quality is provided at a lower bitrate.

[15] The present invention also provides an audio decoding method and apparatus by which the performance of the prior art decoding apparatus is improved such that better sound quality is provided at a lower bitrate.

[16] According to an aspect of the present invention, there is provided a digital audio signal encoding method comprising: (a) based on an input audio signal, generating a time-frequency band table; (b) based on the generated time-frequency band table, searching for a nearest neighbor block of a block being currently encoded, and generating information on the nearest neighbor block; and (c) generating a bitstream containing the generated information on the nearest neighbor block.

[17] According to another aspect of the present invention, there is provided a digital audio signal encoding method comprising: (a) based on an input audio signal, generating a time-frequency band table; (b) based on the generated time-frequency band table, searching for a nearest neighbor block of a block being currently encoded; (c) based on the nearest neighbor block searched for, determining whether or not a block being currently encoded is a redundant block; and (d) based on the result determined in step (c), generating an output bitstream.

[18] According to still another aspect of the present invention, there is provided a digital audio signal encoding apparatus comprising: a time-frequency band table generation unit which, based on an input audio signal, generates a time-frequency band table; a nearest neighbor block searching and nearest neighbor block information generation unit which, based on the

generated time-frequency band table, searches for a nearest neighbor block of a block being currently encoded, and generates information on the nearest neighbor block; and a bitstream packing unit which generates a bitstream containing the generated information on the nearest neighbor block.

[19] According to yet still another aspect of the present invention, there is provided a digital audio signal encoding apparatus comprising: a time-frequency band table generation unit which, based on an input audio signal, generates a time-frequency band table; a nearest neighbor block searching unit which, based on the generated time-frequency band table, searches for a nearest neighbor block of a block being currently encoded; a redundant block decision unit which, based on the nearest neighbor block, determines whether or not a block being currently encoded is a redundant block; and a bitstream generation unit which, based on the result determined in the redundant block decision unit, generates an output bitstream.

[20] According to a further aspect of the present invention, there is provided a decoding method for decoding an audio signal containing additional information on a predetermined region of the audio signal, comprising: (a) decoding a block which is not included in the predetermined region, from an input audio bitstream; (b) based on the decoded block data, generating a time-frequency band table corresponding to the predetermined region; and (c) by using the generated time-frequency band table, reconstructing a current block included in the predetermined region, based on the additional information on the predetermined region of the audio signal.

[21] According to an additional aspect of the present invention, there is provided a decoding method for decoding a digital audio signal comprising: (a) extracting nearest neighbor block information from an input audio bitstream; (b) based on the input audio bitstream, generating a time-frequency band table; (c) based on the extracted nearest neighbor block information, determining whether or not a block being currently decoded is a redundant block; and (d) if the block being currently decoded is a redundant block, by using the generated time-frequency band table, reconstructing the redundant block, based on the extracted nearest neighbor block information.

[22] The method may also comprise reconstructing an entire spectrum corresponding to the input audio bitstream by using the reconstructed redundant block.

[23] According to an aspect of the present invention, there is provided a decoding apparatus for decoding an audio signal containing additional information on a predetermined region of the audio signal, comprising: a decoding unit which decodes a block which is not included in the predetermined region, from an input audio bitstream; and a post-processing unit which, based on the decoded block data, generates a time-frequency band table corresponding to the predetermined region, and by using the generated time-frequency band table, reconstructs a current block included in the predetermined region, based on the additional information on the predetermined region of the audio signal.

[24] According to another aspect of the present invention, there is provided a decoding apparatus for decoding a digital audio signal comprising: a nearest neighbor block information extracting unit which extracts nearest neighbor block information from an input audio bitstream; a time-frequency band table generation unit which, based on the input audio bitstream, generates a time-frequency band table; and a redundant block reconstruction unit which, based on the extracted nearest neighbor block information, determines whether or not a block being currently decoded is a redundant block, and if the block being currently decoded is a redundant block, by using the generated time-frequency band table, the redundant block reconstruction unit reconstructs the redundant block, based on the extracted nearest neighbor block information.

BRIEF DESCRIPTION OF THE DRAWINGS

[25] The above objects and advantages of the present invention will become more apparent by describing in detail exemplary embodiments thereof with reference to the attached drawings in which:

[26] FIG. 1 is a block diagram of a prior art MPEG audio encoding apparatus;

[27] FIG. 2 is a diagram for explaining a spectrum band replication method;

[28] FIG. 3 is a diagram of an encoding apparatus according to an exemplary embodiment of the present invention;

[29] FIG. 4 is a diagram showing a time-frequency band table which is used in the present invention;

[30] FIG. 5 is a flowchart of the steps performed by an encoding method according to an exemplary embodiment of the present invention;

[31] FIG. 6 is a diagram of an encoding apparatus according to another exemplary embodiment of the present invention;

[32] FIG. 7 is a flowchart of the steps performed by an encoding method according to another exemplary embodiment of the present invention;

[33] FIG. 8 is a diagram of a decoding apparatus according to an exemplary embodiment of the present invention;

[34] FIG. 9 is a flowchart of the steps performed by a decoding method according to an exemplary embodiment of the present invention;

[35] FIG. 10 is a diagram of a decoding apparatus according to another exemplary embodiment of the present invention; and

[36] FIG. 11 is a flowchart of the steps performed by a decoding method according to another exemplary embodiment of the present invention.

DESCRIPTION OF THE EXEMPLARY EMBODIMENTS

[37] Voice codecs and video codecs use time correlation between signal samples in order to compress data. Voice codecs use a linear prediction coefficient method to perform compression. Meanwhile, video codecs use motion measuring to perform time correlation.

[38] In general, using time correlation for compressing data is not appropriate for audio codecs, since the characteristics of an audio signal are dynamic and have less time correlation. However, in a frequency transform domain, each subband data signal is static in essence compared to those in a

time domain. Accordingly, the linear prediction method using correlation between frames is used in the frequency transform domain.

[39] For example, in order to achieve a better compression ratio, the MPEG-2 AAC performs linear prediction for each transform coefficient. Also, in order to remove long term periodicity, the MPEG-4 AAC uses a long term predictor which is similar to the linear prediction method.

[40] Referring to FIG. 2, a spectrum band replication (SBR) method using similarity of spectrum coefficients will now be explained.

[41] The SBR method improves performance of an audio and voice codec at a low bitrate, by increasing an audio band at a given bitrate, or by improving encoding efficiency at a given quality level.

[42] According to the SBR method shown in FIG. 2, an encoder does not encode the high frequency part of a frequency spectrum and encodes only the low frequency part, and then transmits the signal. Then, when the signal is decoded, the high frequency part that is not transmitted is reconstructed based on the spectrum of the low frequency part.

[43] For example, in the prior art encoding method, an MP3 encoder employing the SBR method encodes part of a music signal from 0 to 8 kHz. The MP3 file in which only the part from 0 to 8 kHz is encoded can be decoded by a prior art decoder. Therefore, the SBR method is compatible with the prior art MP3. In the SBR method, in order to reconstruct the high frequency part, that is, the part ranging from 8 kHz to 16 kHz, the harmonic

structure of the spectrum is used and also the decoded signal from 0 to 8 kHz is used.

[44] When the SBR method is employed, the narrow audio bandwidth, which is provided at a low bitrate by a codec using the prior art perceptual encoding method, can be expanded such that an analog FM audio bandwidth (15 kHz) or over can be provided. Also, the SBR method improves the performance of a narrow-band speech codec and, for example, it is possible to provide a dedicated voice channel having a 12 kHz audio bandwidth that is used in multilingual broadcasting.

[45] Though additional encoder information for guiding decoding processing is partially processed in the encoder, most steps of the SBR method are performed in the decoder.

[46] From the technical viewpoint, SBR is a method for efficiently encoding a high frequency signal in an audio compression algorithm. An encoding apparatus employing the SBR method transmits only the low frequency part of a spectrum. The omitted high frequency part is generated in a decoding process in the SBR decoder. Instead of transmitting the high frequency part, the decoder employing the SBR method analyzes the spectrum of the low frequency part transmitted by the encoder and reconstructs the high frequency part.

[47] In order to guarantee accurate reconstruction of the high frequency part, some guidance information is transmitted as a bitstream encoded at a low data rate. As a result, the SBR method enables the entire band of an audio

signal to be encoded at a very low data rate and at the same time provides greatly improved compression efficiency compared to the prior art MP3 encoders.

[48] Thus, the LPC algorithm uses time correlation, while the SBR algorithm uses the frequency correlation of a signal.

[49] An algorithm according to the present invention uses both the time and frequency dependencies of an audio signal at the same time. Referring to FIGS. 3 through 11, exemplary embodiments according to the present invention will now be explained.

[50] FIG. 3 is a diagram of an exemplary embodiment of the present invention.

[51] Referring to FIGS. 3 and 4, an audio encoding method according to an exemplary embodiment of the present invention will now be explained.

[52] The encoding apparatus according to the present invention comprises an encoding unit 310, a time-frame band replication (TFBR) unit 320, and a bitstream packing unit 330.

[53] The encoding unit 310 performs a function similar to the prior art audio encoder, that is, the audio encoder shown in FIG. 1. Accordingly, a detailed explanation on the function of the encoding unit 310 will be omitted. Though the audio encoder shown in FIG. 1 is used in the present embodiment, other audio encoders can also be used.

[54] The TFBR unit 320 comprises a time-frequency band table generation unit 322 and a nearest neighbor block searching and nearest neighbor block information generation unit 324.

[55] The time-frequency band table generation unit 322 divides the data signal, which is MDCT transformed in the encoding unit 310, into N frequency blocks in each frame such that a time-frequency index combination, that is, a time-frequency (TF) band table, shown in FIG. 4, is generated.

[56] Though the MDCT transform is used as the time-frequency transform method in the present embodiment, other time-frequency transform methods may also be used.

[57] In the present embodiment, after the MDCT unit of the encoding unit 310 divides the audio signal into a plurality of bands, each band has a plurality of spectrum coefficients. Though bands having an identical width are used in the present embodiment, bands having a variety of widths may also be used.

[58] In FIG. 4, i is a frame index, and j=0, 1, 2, ..., j-1, j, j+1, ..., N is a frequency block index of a frame. Here, i denotes a current frame in which encoding is performed, and i-1 and i+1 denote the previous frame and the next frame, respectively. Meanwhile, j denotes a frequency band in which encoding is performed, j=0 indicates the first frequency band in a frame, and j also denotes a frequency band of a block which is desired to be encoded at present. Also, j-1 indicates the previous frequency band.

[59] For example, B(i, j) of FIG. 4 indicates a block corresponding to a j-th frequency band in an i-th frame, and the number of spectrum coefficients in each block B(i, j) is identical.

[60] The TFBR method using the TF band table shown in FIG. 4 will now be explained in more detail.

[61] The TFBR method according to the present invention uses both the time correlation between frames and the spectrum similarity between frequency bands. Also, the present invention uses the fact that block B(i, j) has a value similar to the value of one block among the previous blocks. This is based on the following facts.

[62] 1. The spectrum of the high frequency part and that of the low frequency part in a signal have inherent similarity.

[63] 2. Though the entire spectrum of each frame is different, part of the spectrum of a current frame is similar to part of the spectrum of the previous frame.

[64] By using equation 1 below, the nearest neighbor block searching and nearest neighbor block information generation unit 324 searches the previous blocks for a block which is the least different from the current block. Here, the previous blocks include not only j previous blocks in the current frame but also the blocks of a predetermined number of previous frames.

$$D(i,j)=\{\,|B(i,j),\ Ck*B(m,n)|\,\}\ldots\ldots(1)$$

where B(m, n) denotes an n-th block of an m-th frame.

[65] Here, if the m-th frame is a current frame, m=i, and n=0, 1, .., j-1. If the m-th frame is a previous frame, m=i-1, i-2, i-M+1, and n=0, 1, …, N-1. Ck is a set of weighting factors, and k=0, 1, …, K-1.

[66] The nearest neighbor block searching and nearest neighbor block information generation unit 324 determines whether or not block B(i, j) that is currently encoded is included in the high frequency band. If the current block B(i, j) is included in the high frequency band, that is, if j is equal to or greater than a predetermined frequency j_{TH}, the m, n, and k values that minimize the difference between B(i, j) and $C_kB(m, n)$ are obtained. The m, n, and k values that minimize D(i, j) are designated as m_{min}, n_{min}, and k_{min}, respectively. The determined m_{min} and n_{min} are referred to as the index of the block which is the least different from the current block B(i, j).

[67] It is determined in the present embodiment whether or not to search for a nearest neighbor block, according to whether or not the frequency band of the current block B(i, j) is equal to or greater than a threshold frequency j_{TH}, that is, whether or not the current block B(i, j) is included in the high frequency band. However, it may also be determined whether to search for a nearest neighbor block based on whether or not the current block is included in an arbitrary frequency band and time domain.

[68] The function |x,y| used in equation 1 is a distance function. In the present embodiment, the function means Euclidian distance function according to equation 2 below. However, it is possible to selectively use a

nearest neighbor classification method using a weighted Euclidian distance function.

$$|x, y| = \sqrt{\sum_{i=1}^{n} (x_i - y_i)^2} \quad \text{.......(2)}$$

[69] Equation 2 considers an n-dimensional feature space, and shows a geometrical distance between two points x=(x1, x2, x3, ..., xn) and y=(y1, y2, y3, ..., yn).

[70] The nearest neighbor block searching and nearest neighbor block information generation unit 324 searches for a block having the least distance among the blocks of the previous frame and the previous blocks of the current frame, by using equation 3 below. The nearest neighbor block determined by the nearest neighbor block searching unit 324 is referred to as $B(m_{min}, n_{min})$.

[71] D(i, j) of equation 1 is the Euclidian distance between the i, j-th block and a block nearest to the i, j-th block, that is, the Euclidian distance between B(i, j) and B_{min} (m_{min}, n_{min}).

[72] D_{min} (i, j), which has the minimum value among the D(i, j) values obtained by equation 1 is presented in equation 3 below.

$$D_{min}(i,j) = | B(i,j), Ck_{min} * B(m_{min}, n_{min}) | \quad \text{......(3)}$$

[73] The bitstream packing unit 330 outputs to the decoder a bitstream containing index information m_{min}, n_{min}, and k_{min} of the nearest neighbor block, that is, a TFBR bitstream, instead of spectrum information on the block B(i, j). Here, only part of the audio signal corresponding to the frequency

band less than j_{TH}, is encoded and included in the output bitstream, and the part equal to or greater than j_{TH} is not included in the bitstream.

[74] When a scale factor is not used in searching for a nearest neighbor block, only index information m_{min} and n_{min} are included.

[75] In the present embodiment, in an MPEG bitstream, the nearest neighbor block index information is included in a field called ancillary data 1. However, the information may be selectively included in fields other than the bitstream.

[76] Also, though the objects of searching for a nearest neighbor block are previous blocks in the present embodiment, it may also be possible to selectively search succeeding blocks for a nearest neighbor block.

[77] FIG. 5 is a flowchart of an audio encoding method according to an exemplary embodiment of the present invention.

[78] In step 510, an audio signal is input and an MDCT which is performed in the prior art audio encoding step is performed on the input time-domain audio signal.

[79] In step 520, the data signal, which underwent MDCT in step 510, is divided into N frequency blocks in each frame and the time-frequency index combination shown in FIG. 4, that is, the time-frequency band table, is generated. Though the MDCT transform is used as the time-frequency band transform method in the present embodiment, other time-frequency transform methods may also be used selectively.

[80] In step 530, it is determined whether or not the frequency of the current block B(i, j) is equal to or greater than the threshold frequency j_{TH}. The threshold frequency j_{TH} is a threshold frequency value for distinguishing a low frequency part from a high frequency part. If the current block is included in the high frequency band, step 540 is performed, and if it is included in the low frequency band, step 550 is performed.

[81] Though in the present embodiment it is determined whether or not the current block B(i, j) is included in the high frequency band, it may also be determined whether or not the block is included in an arbitrary frequency band and time domain.

[82] In step 540, based on the time-frequency band table generated in step 520, a block $B(m_{min}, n_{min})$ nearest to the current block B(i, j) is searched for in the previous blocks of the current block, and the nearest neighbor block information on the nearest neighbor block $B(m_{min}, n_{min})$ is generated. The nearest neighbor block information includes index information m_{min}, n_{min} of $B(m_{min}, n_{min})$. Selectively, when a scale factor is used in searching for a nearest neighbor block, the nearest neighbor block information includes the scale factor k_{min}.

[83] In step 550, the current block included in the low frequency band is encoded.

[84] In step 560, a bitstream, that is, a TFBR bitstream, which includes the nearest neighbor block information, that is, the index information m_{min}, n_{min}, and k_{min} of the nearest neighbor block, which is generated instead of high

frequency band data in step 540 and the current block data encoded in step 550, is generated and output.

[85] FIG. 6 is a diagram of an audio encoding apparatus according to an exemplary embodiment of the present invention.

[86] Referring to FIGS. 6 and 4, the audio encoding apparatus according to an exemplary embodiment of the present invention will now be explained.

[87] The audio encoding apparatus according to the present invention comprises an encoding unit 610, a TFBR unit 620, and a bitstream packing unit 630.

[88] The TFBR unit 620 comprises a TF band table generation unit 622, a nearest neighbor block searching unit 624, and a redundant block decision unit 626.

[89] Since the encoding unit 610, the TF band table generation unit 622, the nearest neighbor block searching unit 624, and the bitstream packing unit 630 perform the same functions as those of corresponding modules in FIG. 3, a detailed explanation thereof will be omitted.

[90] Based on the nearest neighbor block $B(m_{min}, n_{min})$ found in the nearest neighbor block searching unit 624, the redundant block decision unit 626 determines whether or not the current block B(i, j) is a redundant block.

[91] D(i, j) of equation 1 means the Euclidian distance between the current block and a block nearest to the current block, that is, the Euclidian distance between B(i, j) and B_{min} (m_{min}, n_{min}).

[92] D_{min} (i, j), which has the minimum value among the D(i, j) values obtained by equation 1 is presented in equation 3 below.

$$D_{min}(i,j)= \left| B(i,j), Ck_{min}*B(m_{min}, n_{min}) \right| \ldots\ldots(3)$$

[93] If D_{min} (i, j) is less than the threshold Tj, the redundant block decision unit 626 determines that the current block B(i, j) is a redundant block, and transmits the index information m_{min}, n_{min}, and k_{min} of the nearest neighbor block, which is determined in the nearest neighbor block searching unit 624, to the bitstream packing unit 630. Here, the threshold Tj is a threshold in frequency band j, and is an experimentally determined value. In the present embodiment, in an MPEG bitstream, the nearest neighbor block index information is included in the ancillary data 1 field. However, the information may be included selectively in fields other than the bitstream.

[94] Using the nearest neighbor block index information transmitted by the redundant block decision unit 626, the bitstream packing unit 630 outputs to the decoder a bitstream containing index information m_{min}, n_{min}, and k_{min} of the nearest neighbor block, that is, a TFBR bitstream, instead of spectrum information on the block B(i, j).

[95] FIG. 7 is a flowchart of the steps performed by an audio encoding method according to another exemplary embodiment of the present invention.

[96] In step 710, a time-frequency transform such as an MDCT which is performed in the prior art audio encoding step is performed on an input time-domain audio signal.

[97] In step 720, the data signal, which is MDCT transformed in step 710, is divided into N frequency blocks in each frame and the time-frequency index combination shown in FIG. 4, that is, the time-frequency band table, is generated. Though the MDCT transform is used as the time-frequency band transform method in the present embodiment, other time-frequency transform methods may also be used selectively.

[98] In step 730, based on the TF band table generated in step 720, previous blocks of the current block are searched and a block (m_{min}, n_{min}) nearest to the current block B(i, j) is determined.

[99] In step 740, by comparing Dmin(i, j), which is the distance obtained by the equation 3, between the current block B(i, j) and the nearest neighbor block B (m_{min}, n_{min}) determined in step 730, with threshold Tj, it is determined whether or not the current block is a redundant block. If $D_{min}(i, j)$ is less than the threshold Tj, step 750 is performed. If $D_{min}(i, j)$ is greater than threshold Tj, step 760 is performed.

[100] In step 750, it is determined whether the current block is a redundant block, and nearest neighbor block information is generated. Also, a bitstream containing index information m_{min}, and n_{min} of the nearest neighbor block, that is, a TFBR bitstream, is generated and output instead of spectrum information on the block B(i, j). Selectively, when a scale factor is used in searching for a nearest neighbor block, the nearest neighbor block information contains a scale factor k_{min}.

[101] In step 760, it is determined that the current block is a normal block, and a bitstream in which current block data is inserted is generated and output.

[102] FIG. 8 is a diagram of an audio decoding apparatus according to an exemplary embodiment of the present invention.

[103] The audio decoding apparatus 800 shown in FIG. 8 comprises a bitstream unpacking unit 810, and a TFBR decoder 820. The TFBR decoder 820 comprises a decoding unit 822 and a redundant block reconstruction unit 824.

[104] The bitstream unpacking unit 810 extracts TFBR parameters from an input TFBR bitstream. The extracted TFBR parameter is input to the redundant block reconstruction unit 824 and the remaining data is input to the decoding unit 822.

[105] If a current block B(i, j) is a normal block, the decoding unit 822 performs a normal audio decoding process. Since the modules forming the decoding unit 822 perform the same functions as those of an ordinary decoder, a detailed explanation thereof will be omitted.

[106] Based on the decoded normal block data and redundant block data input from the redundant block reconstruction unit 824, the decoding unit 822 generates the TF band table shown in FIG. 4.

[107] Using the TFBR parameters input from the bitstream unpacking unit 810, that is, the TF band table generated based on the index m_{min}, and n_{min} of the nearest neighbor block of the redundant block, the redundant block reconstruction unit 824 approximately reconstructs the redundant block. If the

scale factor k_{min} is used when the TFBR encoder unit generates the TFBR parameters, the scale of the nearest neighbor block is adjusted based on the scale factor k_{min} when the redundant block is reconstructed.

[108] If the nearest neighbor block of the redundant block, that is, the nearest neighbor block which is desired to be referred to in order to approximately reconstruct the redundant block, is a redundant block, the block referred to by the nearest neighbor block is used to reconstruct the redundant block.

[109] The redundant block data which is approximately reconstructed in the redundant block reconstruction unit 824 is input to the decoding unit 822.

[110] Using the redundant block data input from the redundant block reconstruction unit 824, the decoding unit 822 reconstructs the entire spectrum and generates an output audio signal. Using the input redundant block data, the decoding unit 822 updates the TF band table and uses the table when next redundant block data is reconstructed.

[111] FIG. 9 is a flowchart of the steps performed by a decoding method according to an exemplary embodiment of the present invention.

[112] In step 910, the TFBR bitstream transmitted from the encoder is unpacked and the TFBR parameters are extracted.

[113] In step 920, based on the extracted TFBR parameters, it is determined whether or not a block B(i, j) desired to be decoded at present is a redundant block. In the present embodiment, if TFBR parameters corresponding to the current block B(i, j) exist, it is determined that the current block B(i, j) is a redundant block. If it is determined that the current block is a redundant

block, step 930 is performed, and if the current block is not a redundant block, step 940 is performed.

[114] In step 930, based on the TFBR parameters, that is, the index m_{min}, and n_{min} of the nearest neighbor block of the redundant block, the redundant block is reconstructed. Also, if the scale factor k_{min} is included in the TFBR parameters, the scale of the nearest neighbor block is adjusted based on the scale factor k_{min}.

[115] In step 940, it is determined that the current block B(i, j) is a normal block and decoding is performed. Also, in step 940, based on the redundant block data which is reconstructed in step 930, and decoded block data, the TF band table shown in FIG. 4 is generated.

[116] In step 950, based on the normal block data decoded in step 940 and the redundant block data reconstructed in step 930, the spectrum is reconstructed, and based on the spectrum an output audio signal is generated.

[117] FIG. 10 is a diagram of a decoding apparatus according to another exemplary embodiment of the present invention.

[118] The audio decoding apparatus 1000 shown in FIG. 10 comprises a bitstream unpacking unit 1010, a decoding unit 1020, and a post-processing unit 1030.

[119] The bitstream unpacking unit 1010 receives the TFBR bitstream generated in the bitstream packing unit 330 of FIG. 3, and extracts TFBR parameters from the bitstream. The extracted TFBR parameters are input to the post-processing unit 1030.

[120] The decoding unit 1020 decodes a bitstream corresponding to the low frequency part that is transmitted by an ordinary audio encoder, for example, an MP3 encoder, and sends this to the post-processing unit 1030.

[121] Based on the decoded low frequency part data which is input from the decoding unit 1020, the post-processing unit 1030 generates the TF band table shown in FIG. 4, and, based on the TFBR parameters m_{min}, and n_{min} that are input from the bitstream unpacking unit 1010, reconstructs a data block corresponding to the high frequency part. Here, if the scale factor k_{min} is included in the TFBR parameters, the scale is adjusted based on the scale factor k_{min}.

[122] Also, based on the reconstructed high frequency block data, the TF band table which is previously generated is updated. The updated TF band table is used when a next high frequency part block is reconstructed.

[123] As a result, since TFBR parameters m_{min}, n_{min}, and k_{min} have much smaller sizes compared to the size of the original block information, a very small number of additional bits are used. Accordingly, while maintaining the existing transmission bitrate, the sound quality can be effectively improved.

[124] In the present embodiment, it is shown that when high frequency part data is not transmitted, the high frequency part data is restored by using the TFBR parameters. However, the present invention may also be applied selectively to an arbitrary frequency band and frame that are not transmitted.

[125] FIG. 11 is a flowchart of the steps performed by a decoding method according to another exemplary embodiment of the present invention.

[126] In step 1110, the TFBR bitstream is unpacked and the TFBR parameters are extracted.

[127] In step 1120, the input low frequency band block data is decoded and the spectrum corresponding to the low frequency part is generated. In the present embodiment, it is assumed that the input bitstream includes only the low frequency band data. However, the present invention may also be applied selectively to a bitstream containing data of any other frequency band.

[128] In step 1130, based on the low frequency part data decoded in step 1120, the TF band table shown in FIG. 4 is generated, and based on the TFBR parameters m_{min}, and n_{min} that are extracted in step 1110 and the low frequency block decoded in step 1120, the data block corresponding to the high frequency part is reconstructed. Here, if the scale factor k_{min} is included in the input TFBR parameters, the scale is adjusted based on the scale factor k_{min}.

[129] In step 1140, by using the blocks of the low frequency part decoded in step 1120 and the blocks of the high frequency part reconstructed in step 1130, the entire spectrum is reconstructed. Also, based on the reconstructed high frequency part block data, the TF band table is updated. The updated TF band table is used when a next high frequency part block is reconstructed.

[130] The present invention is not limited to the exemplary embodiments described above, and it is apparent that variations and modifications by those skilled in the art can be effected within the spirit and scope of the present invention. Particularly, the present invention may be applied to not only the

MPEG-1 layer 3 but also to all audio encoding apparatuses and methods such as MPEG-2 AAC, MPEG-4, and WMA.

[131] The present invention may be embodied in code, which can be read by a computer, on a computer readable recording medium. The computer readable recording medium includes all kinds of recording apparatuses on which computer readable data are stored. The computer readable recording media includes storage media such as magnetic storage media (e.g., ROM's, floppy disks, hard disks, etc.), optically readable media (e.g., CD-ROMs, DVDs, etc.) and carrier waves (e.g., transmissions over the Internet). Also, the computer readable recording media can be scattered on computer systems connected through a network and can store and execute a computer readable code in a distributed mode.

[132] By using the advanced encoding and decoding method and apparatus according to the present invention described above, the transmission bitrate can be reduced without degradation of sound quality compared to the prior art audio codecs, and sound quality can be improved without raising the transmission bitrate.